## Amendments to the Claims

1. (Currently amended) An audio signal processing apparatus in which a waveform of a digital audio signal to be replayed is processed ~~in a waveform thereof~~, the apparatus comprising:

frequency bandwidth expanding means for expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

low-pass filtering means for performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting means for detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

difference data calculating means for calculating data of a difference ~~data~~ between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting means for weighting the data of the difference [data] depending on the interval of time between the two adjacent waveform peaks; and

producing means for producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference ~~data~~.

2-3. (Canceled)

4. (Currently amended) The audio signal processing apparatus of claim ~~2~~ 1, wherein the past data of the low-pass-filtered digital audio signal used ~~in~~ by the difference data calculating means are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency ~~prior to the current data~~.

5. (Currently amended) The audio signal processing apparatus of claim 4, wherein the weighting means is configured ~~so as~~ to weight the difference data depending on both the interval of time and the polarities of the gradients.

6. (Currently amended) The audio signal processing apparatus of claim 4, wherein the producing means is configured ~~so as~~ to add the weighted difference data to the low-pass-filtered digital audio signal.

7. (Currently amended) A method of processing a waveform of a digital audio signal to be replayed ~~in a waveform thereof~~, comprising ~~the~~ steps of:

expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

calculating data of a difference ~~data~~ between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting the data of the difference ~~data~~ depending on the interval of time between the two adjacent waveform peaks; and

producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference ~~data~~.

8. (Canceled)

9. (Currently amended) The processing method of claim ~~8~~ 7, wherein the past data of the low-pass-filtered digital audio signal are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency ~~prior to the current data~~.

10. (Currently amended) The processing method of claim 9, wherein the weighting step ~~is configured so as to weight~~ weights the difference data depending on both the interval of time and the polarities of the gradients.

11. (Currently amended) A computer-readable program used by a computer for processing a waveform of an inputted digital audio signal to be replayed ~~in a waveform thereof~~, the program ~~comprising~~ allowing the computer to functionally realize ~~the~~ steps of:

expanding a frequency bandwidth of the digital audio signal through conversion of a sampling frequency at which the digital audio signal is sampled;

performing low-pass filtering on the digital audio signal expanded in the frequency bandwidth, the low-pass filtering involving a cut-off frequency corresponding to the converted sampling frequency;

detecting an interval of time between two adjacent waveform peaks of the low-pass-filtered digital audio signal, a polarity of a gradient of the waveform changing at each of the two adjacent waveform peaks and the interval of time being detected by measuring the number of times of sampling based on the converted sampling frequency;

calculating data of a difference ~~data~~ between current data of the low-pass-filtered digital audio signal and past data of the low-pass-filtered digital audio signal;

weighting the data of the difference ~~data~~ depending on the interval of time between the two adjacent waveform peaks; and

producing output data based on both the low-pass-filtered digital audio signal and the weighted data of the difference ~~data~~.

12. (New) The audio signal processing apparatus of claim 1, wherein the producing means adds the weighted difference data to the low-pass-filtered digital audio signal.

13. (New) The processing method of claim 9, wherein the producing step adds the weighted difference data to the low-pass-filtered digital audio signal.

14. (New) The computer-readable program of claim 11, wherein the past data of the low-pass-filtered digital audio signal are data sampled prior to sampling the current data by one sampling period of the converted sampling frequency.

15. (New) The computer-readable program of claim 13, wherein the weighting step weights the difference data depending on both the interval of time and the polarities of the gradients.

16. (New) The computer-readable program of claim 13, wherein the producing step adds the weighted difference data to the low-pass-filtered digital audio signal.